A DOUBLE-SIDEBAND LIMITING RADIO-FREQUENCY SPEECH PROCESSOR FOR INCREASING THE INTELLIGIBILITY OF SPEECH SIGNALS OVER NOISY SINGLE-SIDEBAND CHANNELS

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THESIS

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by

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ABSTRACT

The intelligibility of speech transmitted over noisy Single-Sideband channels can be increased by improving the average-to-peak ratio of a speech waveform while generating as few distortion products as possible.

Previous studies have shown the effect of various distortion products on intelligibility and have concluded that an SSB clipping processor is ideal, although somewhat expensive and complex.

A Double-Sideband Limiting processor was constructed and evaluated against a commercially available SSB-Clipper. Experimental evidence is presented to show that the DSB-Limiting scheme performed as well as the SSB-Clipping processor when the corrupting noise was introduced at audio frequencies.

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I. INTRODUCTION

The object of this investigation is to improve the average-to-peak ratio of an audio-frequency speech signal. The processed signal is to be used in a single-sideband communication system to improve intelligibility over noisy channels.

The processing is accomplished by (1) Translating the audio signal into a double-sideband suppressed carrier (DSB) signal at 5.5MHz. (2) Processing this DSB signal with a fast-acting limiting bandpass amplifier and (3) Coherently detecting the processed DSB signal for use as the audio input to the single-sideband communications system.

Although clipping at audio frequencies will improve the average-to-peak ratio, it has been shown that the resulting harmonic and especially intermodulation distortion adversely affect intelligibility and the quality of the sound [Ref. 1].

Repeaking, associated with using audio-frequency clipped signals in a SSB system, acts to reduce the average-to-peak ratio, exactly opposite to the desired effect [Refs. 2, 3, and 4].

The improvement of the average-to-peak ratio using SSB-Clipping within the transmitter at radio frequencies is well documented, [Refs. 2, 4, and 5], and considered to be ideal. For a given level of intelligibility, up to 10 dB improvement in the power level required for successful communications over noisy channels has been reported [Ref. 9]. The use of RF SSB clipping in existing equipment requires extensive circuit modifications. An alternate method is to use an

[illegible]

add-on unit in which the audio input is converted to SSB which is then clipped, filtered and translated back to audio. This processed audio signal is used as the input to a regular SSB transmitter. A commercial unit of this type is available, but apparently has not been evaluated by controlled experiments [Ref. 14]. However, it should be nearly as effective as internal RF SSB clipping because of the low number of distortion products generated, the improvement in average-to-peak ratio achieved and, the experience of radio amateurs using the device who enthusiastically acclaim its effectiveness.

To use SSB clipping either internal circuit modifications are required (for RF Clipping), or bulky or expensive filters are necessary (in an add-on unit).

The proposed DSB-Limiting add-on unit is used for reasons of circuit simplicity and low cost. This device accepts an audio frequency signal, processes it at RF using a DSB-Limiting technique, and returns an audio-frequency signal with much improved average-to-peak characteristics and low harmonic and intermodulation distortion. It can be used with existing equipment merely by routing the audio input signal through the device, no internal circuit changes being required.

Performance should approach that of an SSB processor, any reduction being due to slightly higher IM distortion which will occur. The added advantage of lower cost and simplified circuitry achieved by elimination of the SSB filter should compensate for any slight reduction in performance.

II. PROPERTIES OF SPEECH SIGNALS

Speech may be classified into voiced and unvoiced sounds. The voiced sounds are called vowels and the unvoiced sounds consonants. The transitional sounds between voiced and unvoiced, such as dipthongs and semi-vowels, contain properties common to both vowels and consonants.

A speech sound begins with a flow of air under pressure from the lungs through the trachea to the larynx. At the larynx, the air stream may be periodically interrupted by the vibration of the vocal chords to produce a voiced sound, or it may pass through uninterrupted producing an unvoiced sound. For a voiced sound, the periodic pulses of air excite the resonant modes of the vocal tract cavities. As the tract cavities change by movement of the throat, tongue, jaw, etc., different sounds are produced [Ref. 6].

The fundamental frequency of vibration of the vocal chords varies from about 90 Hz for a man to about 350 Hz for a woman or child [Ref. 1]. The vibration is not sinusoidal as the vocal chords may be closed for as long as half a cycle. Therefore, the sound spectrum will contain many harmonics.

Voiced sound characteristics are derived from the configuration of the vocal tract, throat and oral cavities. Corresponding to a given configuration of the tract, there is a certain set of acoustic resonances which selectively enhance the harmonics of the sound source. These spectral regions of reinforcement are called formants. The nasal tract acts in the same manner, but its configuration is essentially fixed.

The formant regions of accentuated spectral intensity are usually two or three in number. The relative location of these regions is an important characteristic in the difference between vowel sounds [Ref. 1]. Generally, it is necessary to retain these characteristic structures or the sound character of vowels will be lost [Ref. 5]. The formant frequencies do not occur at the same frequency when the same vowel is spoken by different persons. All formant frequencies are higher for a woman and much higher for a child. The relation between the harmonic frequencies in each formant region and the distribution of intensities are characteristics of the vowels, rather than the absolute values of the formant frequencies.

The consonants, or unvoiced sounds, are formed by forcing air through the vocal tract without vibration of the vocal chords. Consequently, consonants do not have the regions of reinforced intensity characteristic of vowels. Consonants have no harmonic relation existing between frequency components. The random-like noise of air flowing through the vocal tract cavities excites various cavity resonances, which generally are not harmonically related.

Frequently, only the air present in the mouth cavity is utilized in producing the consonants so that the intensity of the consonants tends to be much less than that of vowels. The intensity of consonant sounds is concentrated in the high frequency end of the speech spectrum [Ref. 7].

In studies of speech characteristics by Fletcher and others, it has been found that the intelligibility of speech in practically all modern languages is a function of the higher frequency consonant

sounds, although most of the energy is carried in the lower frequency vowel sounds [Ref. 1].

The peak factor for speech has been found to be approximately 14 dB and is defined as follows:

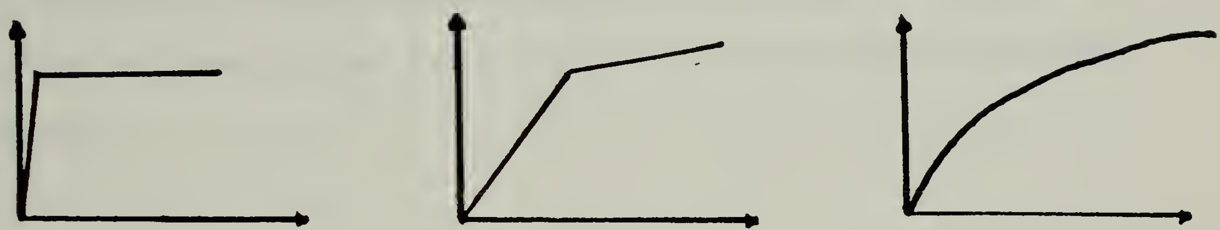
$$PF = 20\log_{10} \frac{\text{peak value}}{\text{rms value}}$$, its spectrum

extends from about 300 to 3000 Hertz [Ref. 7].

In a peak power limited system, as most communication systems are, the best average-signal to average-noise ratio can be achieved with a signal having a high average-to-peak ratio. Unprocessed speech, with a peak factor of 14 dB, makes poor use of available power in communications systems.

III. ELECTRONIC PROCESSING TECHNIQUES

Three techniques for processing signals to limit their peak excursions thus increasing the average-to-peak ratio, are shown in Figure 1.



a. Clipper b. Limiter c. Compressor

horizontal axis: input vertical axis: output

Input-Output Characteristics of Processors

Figure 1

A. CLIPPER

The clipper, Figure 1a., is a fast acting, 'instantaneous' device whose output amplitude is severely clipped at an essentially constant amplitude independent of input amplitude, after a very small input threshold is reached. The device is best described as a 'waveform slicer'.

B. LIMITER

The limiter, Figure 1b., is a device that has a constant slope linear input-output characteristic until a threshold is reached, after which a gain reduction adjustment takes place. This new slope is relatively constant over the remaining dynamic range of the device. The time constants or more commonly called the 'attack' and 'release' times of the limiter can be fast or slow. These times are defined as

the time required for a gain adjustment to be 90% complete. Typically, the attack time for voice limiters is five milliseconds and the release time one-half second or longer.

C. COMPRESSOR

The compressor, Figure 1c., is a device that has a continuously varying input-output characteristic. The gain reduction adjustment is continuously varied over the operating range of the device. The attack and release times are typically slow, on the order of one-half second.

IV. DISTORTION AND INTELLIGIBILITY

Each of the three processing techniques increases the average-to-peak value of the input waveform. Unfortunately, all nonlinear systems produce distortion. Results of many investigations have established the effect of various distortions of speech waveforms on intelligibility. Intelligibility, in this context, refers to a measure of how much of the original information is assimilated by the listener. This implies only that the information is understood and nothing about the naturalness of the speech.

Amplitude, frequency and phase distortion are possible when a signal is processed. The following sections discuss the effect of each of these on the intelligibility of speech.

A. AMPLITUDE DISTORTION

Rather comprehensive tests over many years indicate that amplitude distortion does not drastically reduce intelligibility [Ref. 3]. Even a synthesized waveform using only the zero-crossing information (frequency) in a speech waveform will yield intelligible speech, albeit of low quality. In general, results show that the average power and frequency components are important. The exact waveshape is not [Ref. 8].

B. PHASE DISTORTION

Phase distortion has very little effect on intelligibility as shown by Fletcher [Ref. 6]. He concludes that phase distortion has only a

[illegible]

slight effect on the quality of speech and essentially none on intelligibility.

C. FREQUENCY DISTORTION

Frequency distortion is introduced in any nonlinear processing system in the form of harmonic generation and intermodulation (IM) production. In addition to the original frequencies, new frequencies of sums and differences of the original and their harmonics are produced. Since speech consists of many frequency components, any waveform processing will create pronounced IM and harmonic distortion.

D. INTELLIGIBILITY

The parameters necessary to recognize different words are frequency and intensity. Lannes showed in 1966 that the IM products generated by speech passing through a nonlinear process were the most significant in reducing intelligibility [Ref. 1]. The IM products were treated as noise, thus tend to mask the weaker sounds.

Each of the three techniques of clipping, limiting and compressing can be applied to a speech signal to improve its average-to-peak ratio. Clipping provides the greatest improvement, but generates the greatest amount of distortion. Limiting has less distortion and less improvement. Compressing has the least improvement and the least distortion.

E. AUDIO PROCESSING

For a speech waveform at audio frequency, processing produces both harmonic and IM distortion. If the input signal is a two-tone

signal of the form A $\cos w_1 t$ + B $\cos w_2 t$ the output will contain, in addition to the fundamental and harmonics, intermodulation products given by $nw_1 \pm mw_2$ where m and n are positive integers. It is apparent that audio frequency processors have essentially all distortion products of the fundamental and low order harmonics which fall within the audio spectrum. As speech is composed of many frequencies, the distortion products are much more numerous than in this simple example.

F. SINGLE-SIDEBAND PROCESSING

The single-sideband spectrum of an audio frequency speech waveform consists of each frequency of the original wave added (upper sideband) or subtracted (lower sideband) from the suppressed carrier [Ref. 16]. For example, if two frequencies of 500 and 1200 Hertz are translated to 250kHz, the lower sideband has components 249.5 and 248.8 kHz. If this signal is clipped, assuming a third-order nonlinearity, ($e_{out} = K_1 e_{in} + K_2 e_{in}^2 + K_3 e_{in}^3$,) the output will contain frequency components of 249.5, 248.5, 499.0, 497.6, 797.6, 748.5, 746.4, 250, 500, 750 kHz plus sums and differences. A sharp bandpass filter will remove all components not in the desired range of 247.0 to 249.5 kHz. Thus the remaining components will be 248.8 (original), 249.5 (original) and 248.1 (497.6-249.5). When this wave is translated back to audio, it will contain only one distortion product, compared to the numerous products remaining from an audio processor of the same type. A commercial processor of the SSB-Clipper design is in use and performs well in Amateur Radio Service [Ref. 14].

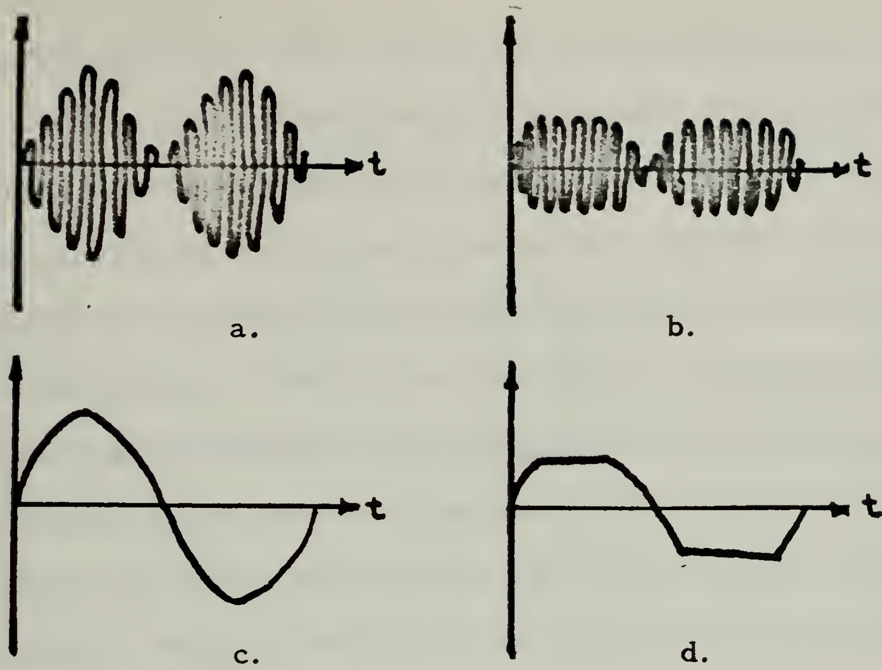
G. DOUBLE-SIDEBAND CLIPPING

The double-sideband spectrum consists of each frequency of the original wave added and subtracted from the suppressed carrier. For example, if two frequencies of 500 and 1200 Hertz are translated to 250 kHz, the spectrum contains 249.5, 248.2, 250.5 and 251.2 kHz.

If this signal is now clipped, again assuming a third order non-linearity, the output will contain frequency components of sums and differences of the original frequencies and harmonics. Many more of these will fall into the passband of a filter following the clipper. A rigorous Fourier analysis of this process is time consuming and produces a not easily interpretable result. A more satisfying approach is to use the time domain method.

Figure 2a. shows a DSB signal containing one modulating frequency. Figure 2c. shows the coherently detected waveform of this signal. It shows that the envelope of the DSB signal taken from above the zero crossing exists for one-half the period of the modulating frequency, and below the zero-crossing axis for the remaining half period is the waveform of the modulating signal.

When the waveform of Figure 2a. is passed through a clipper, the waveform shown in Figure 2b. results. The detected waveform is shown in Figure 2d. It is apparent that DSB clipping is equivalent to audio frequency clipping. Therefore, one would not expect any improvement over audio frequency processing using DSB-Clipping.



Time domain waveforms of DSB signals

Figure 2

H. DSB-LIMITING

Usual limiting action operates with specific 'attack' and 'release' times. The gain reduction of a limiting amplifier is initiated by the peak of the impressed signal, and since, in speech, the greater peaks occur at the lower frequency vowel sounds, it is these vowel sounds that cause the limiting. But, due to the finite time required for gain reduction to take place, parts of these 'action' sounds pass through at approximately full amplitude. The gain reduction then takes place for the rest of the vowels and for the consonants that immediately follow the vowels, as there is insufficient time for a return to normal gain. In addition, the frequent occurrence of these vowel sounds tends to keep the limiter in a continuous state of gain reduction when operating with normally used time constants.

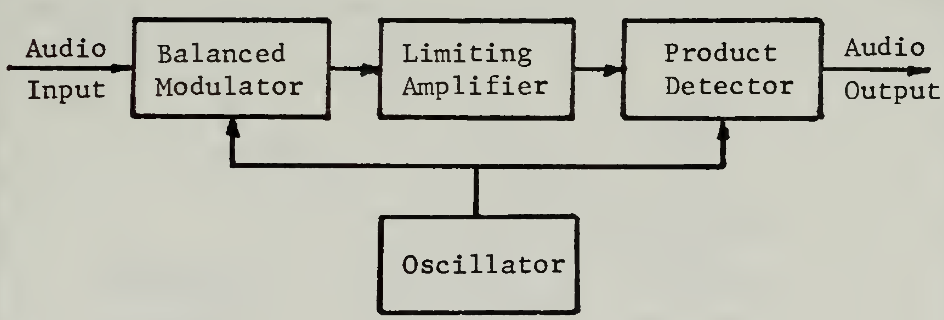
The purpose of this investigation was to design and construct a fast-acting limiter bandpass amplifier operating on a DSB signal. The gain reduction has a time constant such that it will act to reduce the gain during the portion of words or syllables containing intense sounds and recover quickly so that the weaker consonant sounds are fully amplified. This has the dual effect of increasing the average-to-peak ratio and increasing intelligibility by increasing the energy of the important, weaker consonant sounds. The use of DSB results in more IM distortion products than the use of SSB. However, the bandpass amplifier will reduce the distortion to a much lower value than for a comparable audio frequency processor.

V. CIRCUIT DESCRIPTION AND DESIGN

A. GENERAL

Along with demonstrated performance in processing speech signals, the object of this investigation was circuit simplicity and low cost. Therefore, integrated circuits (IC's) were used wherever possible.

As shown in Figure 3, the audio input signal is fed to an IC balanced modulator where a DSB signal is generated. The signal is capacitively coupled to an IC fast-acting limiting bandpass amplifier whose output is the processed DSB signal which is translated back to audio by an IC product detector. This audio signal is the processed audio output. A descrete transistor oscillator-buffer provides the injection carrier to both the balanced modulator and product detector.

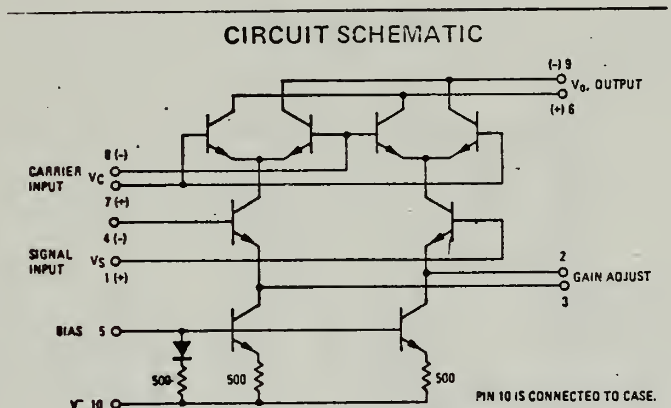


DSB Processor Block Diagram

Figure 3

B. DETAILED DESCRIPTION

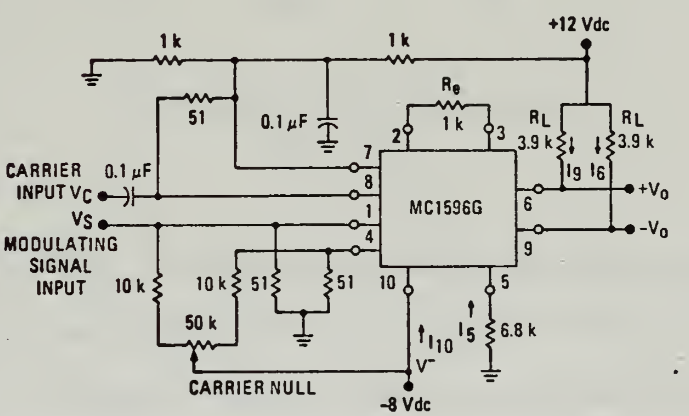
A Motorola type MC1596G monolithic balanced modulator-demodulator is used in both the balanced modulator and product detector. It is a silicon expitaxial passivated device with a balanced input and output with carrier suppression between -50 to -65 dB. It can be used at frequencies up to 10 megahertz. Figure 4 is the schematic diagram of the device.



Motorola type MC1596G IC

Figure 4

This integrated circuit is wired into the circuit as shown in Figure 5.



Balanced Modulator Wiring

Figure 5

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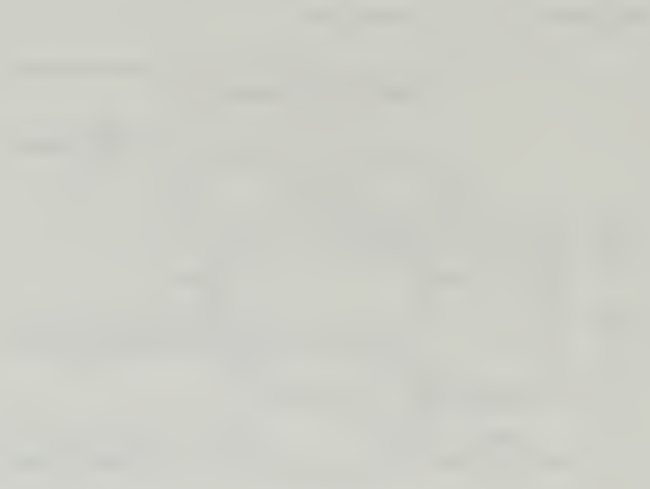

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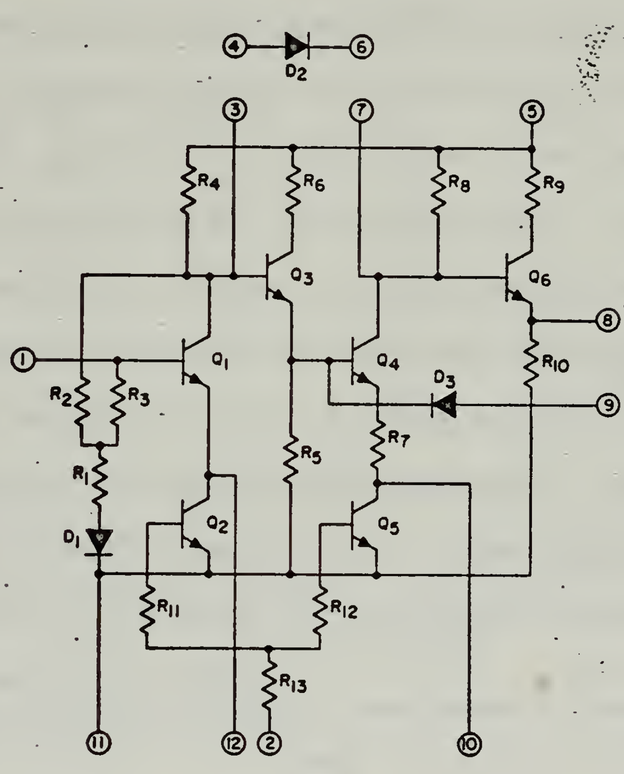
Carrier injection is made adjustable by means of a 5-25pf variable capacitor at the output of the oscillator buffer as shown in Figure 11. This is adjusted to provide the optimum injection for the lowest distortion.

The output of the balanced modulator is capacitivly coupled to the RCA type CA3023 high gain RF amplifier IC. Figure 6 is the schematic of this device.



RCA type 3023 IC Schematic
Figure 6

This IC has a closed loop gain of 40 dB and is capable of gain adjustment by means of a voltage applied to terminal 2. For stable operation, a feedback network must be connected between terminals 3 and 7. Used as a bandpass amplifier, a tuned circuit is connected between the feedback terminals. This provides the greatest DC

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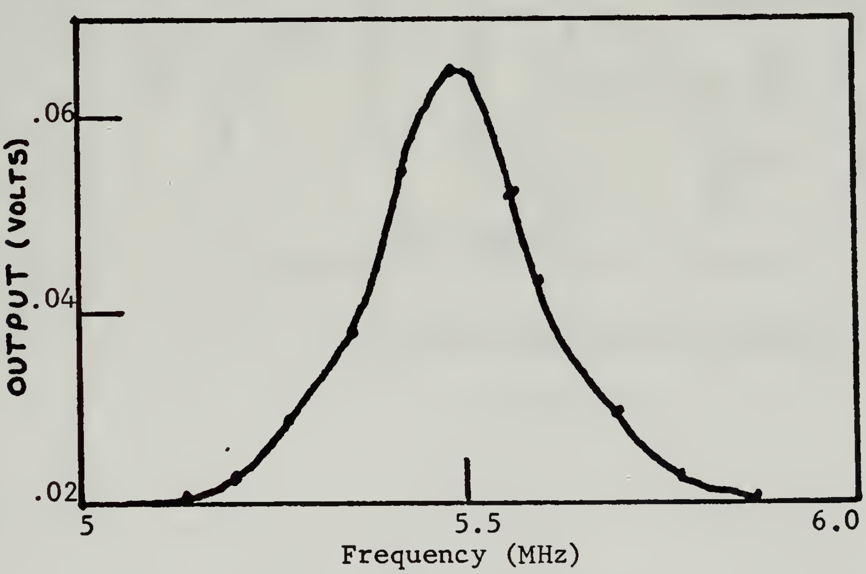
[illegible]

[illegible]

stability and permits AC gains of 50dB at the resonant frequency of the tuned circuit.

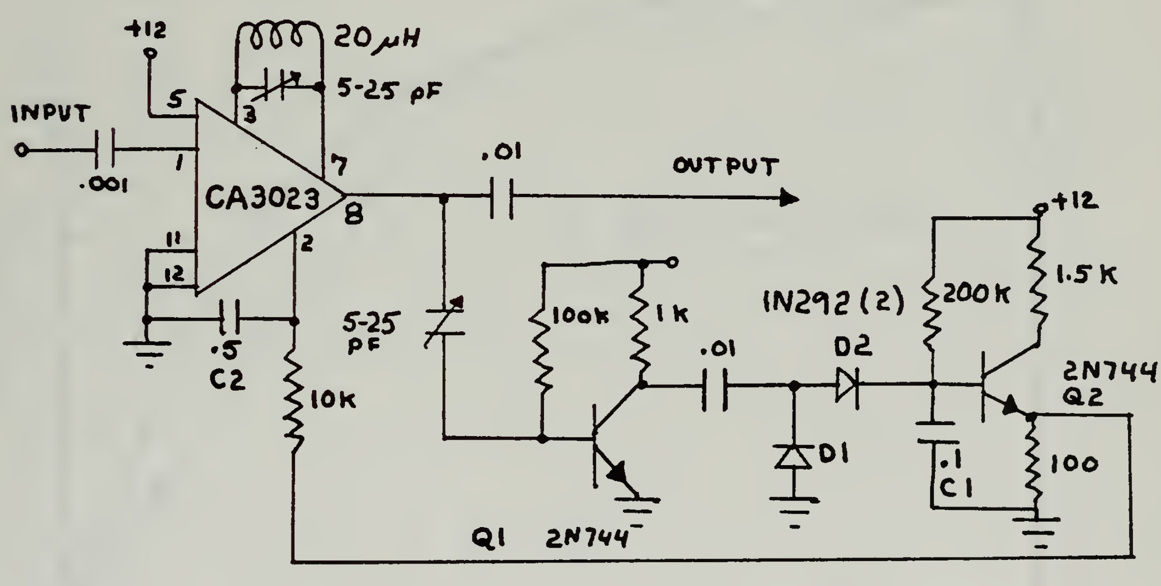
This stage is operated as a fast-acting limiting amplifier and also a bandpass filter, removing distortion products generated by the limiting action. An operating frequency of 5.5 MHz was chosen. This is well within the capabilities of the integrated circuits. It is sufficiently high to provide a narrow bandpass characteristic using only one tuned circuit in the feedback network of the amplifier IC.

For this feedback circuit, a 40 microhenry coil was wound on a ferrite toroid, type T 30-2. A toroid was chosen because the field is confined essentially to the interior of the coil. This minimizes coupling to other parts of the circuit and possible mutual interference with other nearby equipment operating near this frequency. The measured unloaded Q was 170. A 5-25pf variable capacitor was used to resonate the coil to the operating frequency. Figure 7 is a curve of the overall bandpass characteristic of the amplifier. The total effective Q is 25. The output of the amplifier is 50dB down at the nearest harmonic of 11 MHz.



Bandpass Amplifier Response
Figure 7

A small portion of the output of the amplifier is fed to buffer amplifier Q1 as shown in Figure 8. The amplified signal is rectified by voltage doubler D1 and D2. The DC, which is proportional to the DSB level, is amplified by Q2 and applied to the gain control terminal of the IC amplifier. The filter consisting of C1 and C2, along with the 7.5K internal driving point impedance of the gain control terminal, establishes the gain time constant of the system. The values were chosen so that the gain adjustment was 99 percent complete within one millisecond of an applied level change. The release and attack times are the same in this circuit.

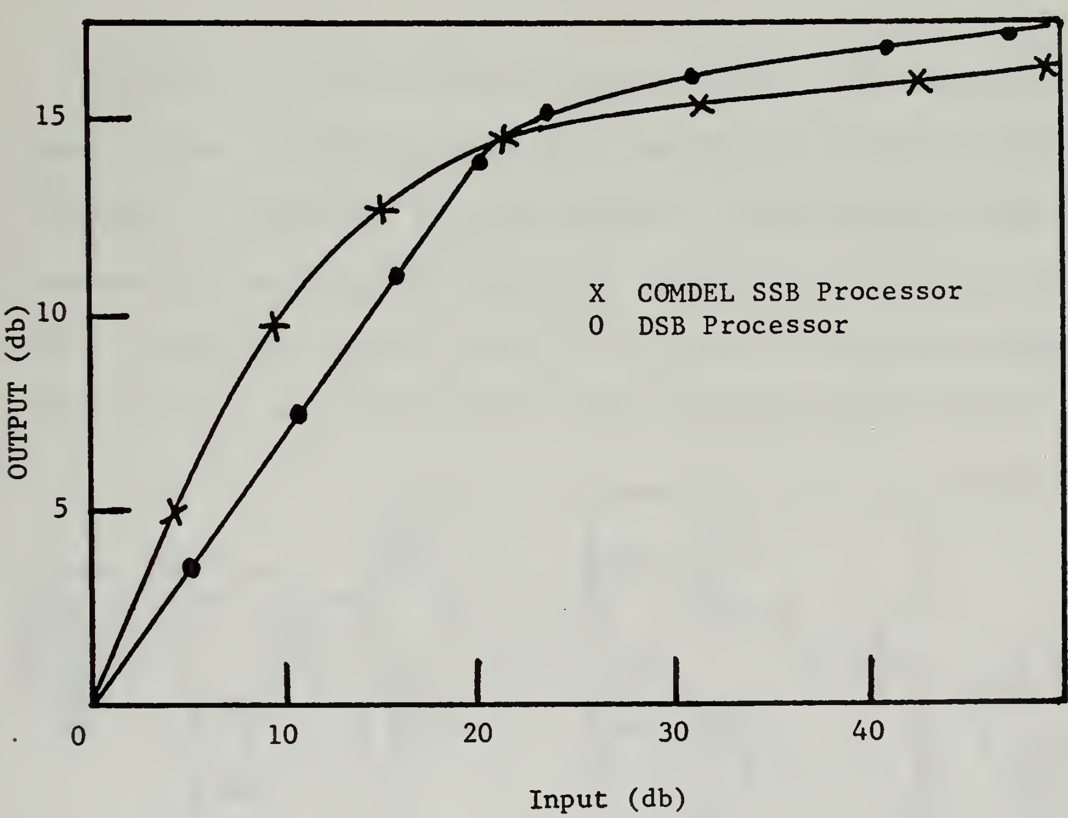


Limiter Bandpass Amplifier

Figure 8

Figure 9 shows the resulting input-output characteristic obtained plotted with the commercial COMDEL SSB processor for comparison. It can be seen that the circuit provides relatively constant gain until the input reaches approximately 20dB, at which point a gain reduction takes place, limiting the output for the remaining useful dynamic range. The useful dynamic range is considered to be that range of input signals for which the output signals contain 10 per cent or less distortion products. It can be seen that for an input variation of 45 dB, the output variation is 16 dB, a 29 dB improvement.

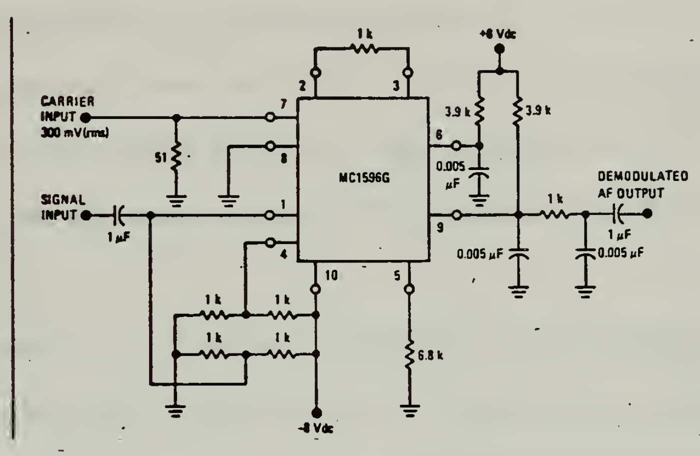
The output of the RF limiter-amplifier is a filtered DSB signal. This is detected by the second Motorola type MC1596G IC.



Input-Output Characteristics

Figure 9

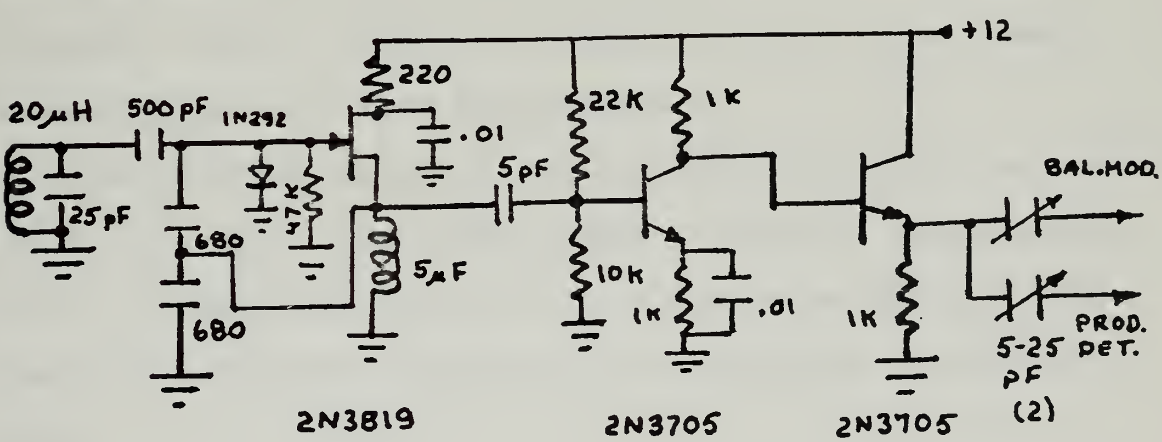
The product detector is wired as shown in Figure 10. As with the balanced modulator, a variable capacitor is used to adjust the carrier injection for the lowest distortion.



Product Detector Wiring

Figure 10

Figure 11 is the oscillator-buffer circuit. The oscillator also uses a Type T 3022 toroid for the resonant circuit for the same advantages as mentioned for the amplifier. The oscillator output is buffered and amplified by transistors Q1 and Q2. Great care was used to generate as pure a sine wave as possible so that minimum distortion is created in the balanced modulator-demodulator.



Oscillator-Buffer Diagram

Figure 11

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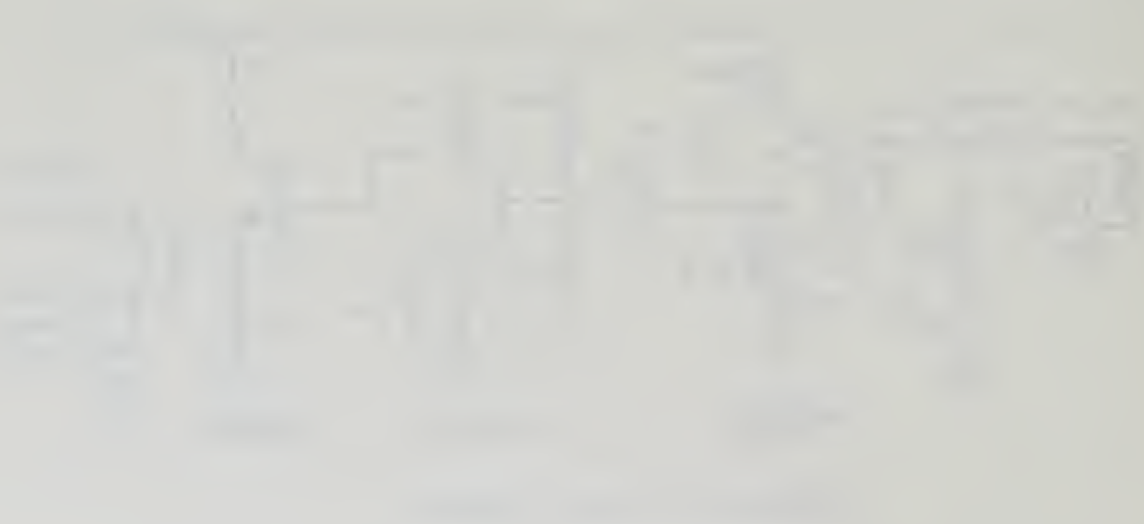

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VI. EXPERIMENTAL PROCEDURE AND RESULTS

A. INTELLIGIBILITY MEASUREMENTS

The most commonly accepted method of testing the intelligibility of a speech processing system is the articulation test. Listeners hear a selected list of sounds, words, or sentences and record what they hear.

The results are compared with the list actually transmitted through the system under test and a mean score computed. This is compared against known scores achieved using other systems to determine the relative merits of the system under test [Ref. 17].

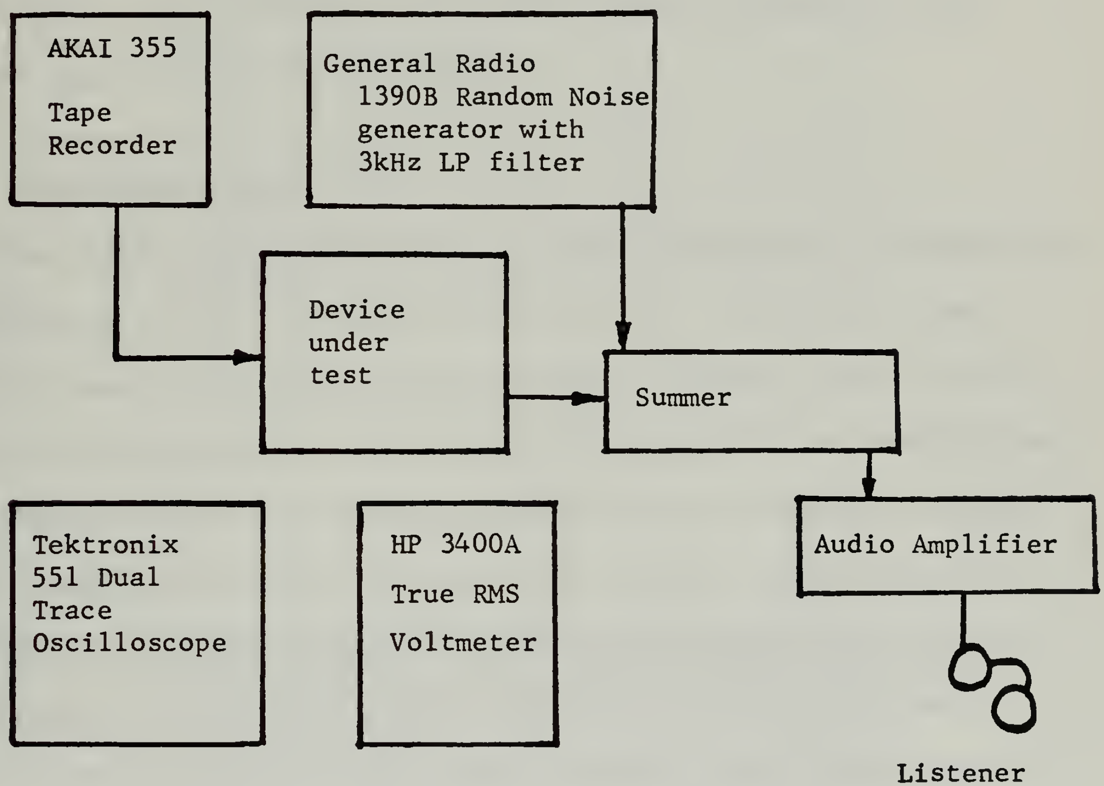
The test results shown in this section were obtained by use of phonetically-balanced word lists. These are lists in which speech sounds occur with approximately the same frequency as they occur in the English language, and the words are so chosen that they are neither very easy nor very difficult.

Word lists, rather than sentence lists or sound lists, were used because sound lists require very careful speakers and trained listeners, neither of which were available. Sentence lists require excessive time to give and compute grades.

The word lists were recorded on an AKAI model 355 tape recorder. Each word was preceded by the preparatory phrase: 'the next word is ______.' This procedure was used to prepare the listener for the test word and to help keep the voice level even while recording the lists.

B. EXPERIMENTAL SET UP

Figure 12 shows the experimental set up used in tests conducted. Both a true rms voltmeter and a dual trace oscilloscope were used to monitor the input and output levels.



Experimental Set Up

Figure 12

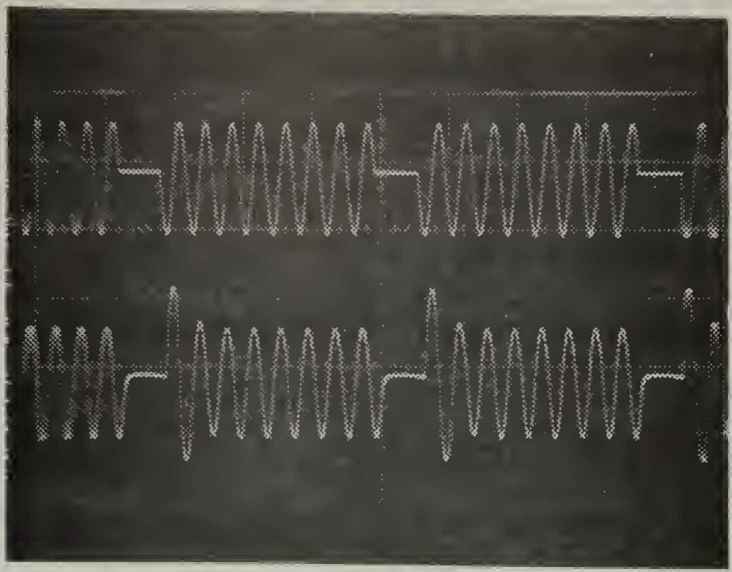
The operation of the DSB-Limiting processor was observed using a Wavetek Model 116 Tone Burst Generator as the input. No noise was added. A one-KHz tone burst was chosen. The input and output signals were simultaneously observed on the Tektronix Model 551 Dual Trace Oscilloscope. Figure 13 is a photograph of this test. It shows that the gain adjustment is essentially complete within one cycle (one-millisecond) after application of the tone burst. The sine

wave itself is visually free of distortion, supporting the belief that DSB-Limiting is a valid technique.

Figure 14 is a oscilloscope photograph of the word SOAP. The decrease of the peak portions of the wave is apparent. Especially noteworthy is the relative increase in the weak consonant sounds before and after the strong vowel $\bar{0}$.

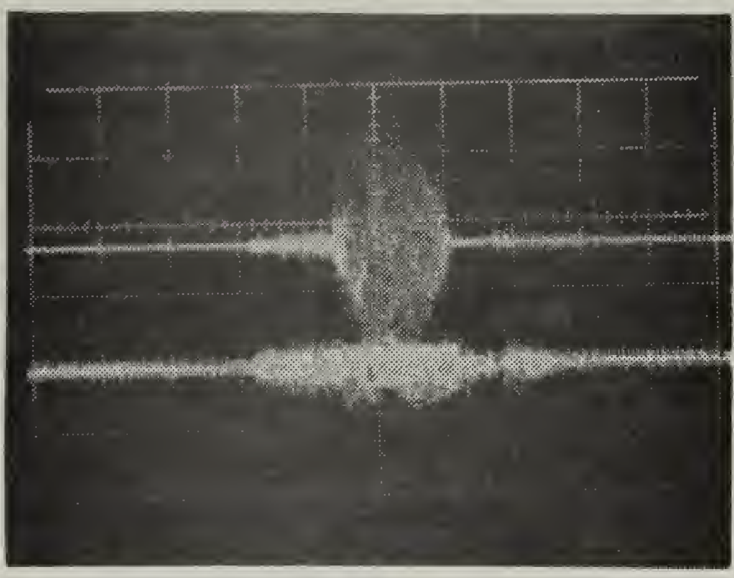
C. LISTENING TESTS

The following intelligibility tests were performed to compare the COMDEL to the DSB-Limiting processor when the output of these devices was corrupted with audio-frequency additive noise. This is not exactly the same as feeding the processed audio into an SSB communication system and adding the noise at the RF channel frequency. But, if a perfect product detector were available, signal plus band-limited noise at RF translated to audio is equivalent to an RF signal coherently detected and summed with audio-frequency low-pass filtered noise [Ref. 16]. The degradation in signal-to-noise ratio is slight for practical double balanced product detectors such as used here. Therefore, the procedure used is valid in evaluating the reportedly good processor (SSB) with respect to the DSB-Limiting device. Both the COMDEL and DSB-Limiter were in the "Device Under Test" section of the block diagram shown in Figure 12. Levels were adjusted so that the output of the tape recorder and the input to the summer could be switched from unit to the other with no change in either impedence or voltage level. The signal-to-noise ratio was adjusted so that the subjects made errors in correctly identifying words about fifty percent of the time. This occurred with a rms noise



1 kHz tone burst. Top trace input,
bottom trace processed output

Figure 13



Spoken word "SOAP" Top trace input,
bottom trace processed output

Figure 14

voltage of 0.12 volts and peak audio voltage of 0.25 volts at the output of the summer. This ratio was maintained for all intelligibility tests.

Each subject was carefully briefed as to what he should listen for. Then, a non-graded familiarity run was conducted. The subject could adjust the audio amplifier gain for the most comfortable volume in the phones. After each subject felt he was ready, two graded runs of 30 words each were made. Two phonetically-balanced word lists were used. One list was played through the COMDEL and the other through the DSB-Limiter. The order of the devices presented and the word list used were carefully balanced so that one-half of the subjects heard the COMDEL first. The word lists were alternated with each subject, no subject hearing the same list twice. The results of twenty-five tests each on the COMDEL and DSB-Limiter conclusively show that the DSB processor performs as well as the COMDEL in the presence of additive noise at audio frequencies. The mean and standard variation of the results is shown in Table I.

Processor	Mean	Variance
COMDEL	14.2	51.2
DSB	14.25	44.79

Table I. Statistical Results of Articulation Tests. Thirty words each test.

VII. CONCLUSIONS AND RECOMMENDATIONS FOR FURTHER WORK

This investigation has shown that the use of a double-sideband limiting technique to improve the average-to-peak ratio of an audio frequency speech signal is as effective, within experimental measurement accuracy, as the more expensive and complex single-sideband technique, when the noise is added at audio frequency.

It is recommended that confirming tests be made on both units using the outputs to drive a single-sideband communications circuit where noise can be injected at the RF Channel frequency.

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13. ABSTRACT

The intelligibility of speech transmitted over noisy Single-Sideband channels can be increased by improving the average-to-peak ratio of a speech waveform while generating as few distortion products as possible.

Previous studies have shown the effect of various distortion products on intelligibility and have concluded that an SSB clipping processor is ideal, although somewhat expensive and complex.

A Double-Sideband Limiting processor was constructed and evaluated against a commercially available SSB-Clipper. Experimental evidence is presented to show that the DSB-Limiting scheme performed as well as the SSB-Clipping processor when the corrupting noise was introduced at audio frequencies.

KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Speech processor						
Double sideband						
Limiting						